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09/891,876	06/26/2001	Shingo Kiuchi	9333/274	9050

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EXAMINER	
VO, HUYEN X	
ART UNIT	PAPER NUMBER
2655	6

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No. 09/891,876	Applicant(s) KIUCHI ET AL.	
	Examiner Huyen Vo	Art Unit 2655	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *12 July 2000* .

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-18* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-18* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *12 July 2000* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

11)☐ The proposed drawing correction filed on _____ is: a)☐ approved b)☐ disapproved by the Examiner.

If approved, corrected drawings are required in reply to this Office action.

12)☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. §§ 119 and 120

13)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____ .

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

14)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).

a) ☐ The translation of the foreign language provisional application has been received.

15)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO-1449) Paper No(s) _____ .
4) ☐ Interview Summary (PTO-413) Paper No(s). _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: .

DETAILED ACTION

Claim Rejections - 35 USC § 102

The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –(b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

Claims 1-2, 8-9,12-13, and 16 are rejected under 35 U.S.C. 102(b) as being anticipated by Arslan et al. (US. Patent No. 6,263,307).

1. Referring to claim 1, Arslan et al. discloses a voice feature extraction device comprising:

a noise reduction system coefficient calculation unit that calculates beforehand a noise reduction system coefficient of a noise reduction system to be used (col. 6, ln. 3-5), and

an input voice power spectrum calculation unit that calculates a power spectrum vector of a processed input voice (904 of figures 9a or 9b), wherein

the noise reduction system that is set to the coefficient calculated by the noise reduction system coefficient calculation unit executes an operation processing the power spectrum vector calculated by the input voice power spectrum calculation unit (figure 9a or 9b).

2. Referring to claim 2, Arslan et al. discloses a voice feature extraction device, wherein the noise reduction system coefficient calculation unit includes a filter coefficient calculation unit that determines a filter coefficient of the noise reduction system to be used, and a power calculation unit that converts the filter coefficient acquired by the filter coefficient calculation unit into the power spectrum vector (col. 14, ln. 30-48).

3. Referring to claim 8, Arslan et al. discloses a voice feature extraction device comprising:

a noise reduction system coefficient calculation unit that calculates beforehand a noise reduction system coefficient of a noise reduction system to be used (col. 6, ln. 3-5),

a microphone that receives the voice of a user (figures 1a or 1b),

a window function operation unit that samples a voice signal received by the microphone, and prevents generation of high frequency components caused by a data jump at intervals of each frame (figures 3 or 4),

an input voice power spectrum calculation unit that calculates a power spectrum vector of the input voice signal processed by the window function operation unit (904 of figures 9a or 9b), and

a noise reduction system that sets the power spectrum vector calculated by the input voice power spectrum calculation unit to the coefficient calculated by the noise

reduction system coefficient calculation unit, and executes an operation processing (figures 9a or 9b).

4. Referring to claim 9, Arslan et al. discloses voice feature extraction device, wherein the noise reduction system coefficient calculation unit includes a filter coefficient calculation unit that determines a filter coefficient of the noise reduction system to be used, and a power calculation unit that converts the filter coefficient determined by the filter coefficient calculation unit into the power spectrum vector (col. 14, ln. 30-48).

5. Referring to claim 12, Arslan et al. discloses a method of extracting voice features comprising:

calculating in advance a noise reduction system coefficient of a noise reduction system to be used (col. 6, ln. 3-5), and

calculating a power spectrum vector of a processed input voice (904 of figures 9a or 9b),

wherein the noise reduction system having the calculated noise reduction system coefficient set to the power spectrum vector (figure 9b), and extracts the voice features (LPC subsystem of figure 4).

6. Referring to claim 13, Arslan et al. discloses a method of extracting voice features, wherein the noise reduction system coefficient is calculated by determining a

filter coefficient of the noise reduction system to be used, and by converting the determined filter coefficient into the power spectrum vector (col. 14, ln. 30-48).

7. Referring to claim 16, Arslan et al. discloses a method of extracting voice features comprising:

calculating in advance a noise reduction system coefficient of a noise reduction system to be used (col. 6, ln. 3-5),

sampling an input voice signal received by a microphone (figures 1a or 1b),

executing processing to prevent generation of high frequency components of the input voice signal sampled (figures 3 or 4),

calculating a power spectrum vector of the signal that is processed to prevent generation of high frequency components (904 of figures 9a or 9b), and

calculating a voice feature from the power spectrum vector by means of the noise reduction system having the calculated noise reduction system coefficient set (figure 9b).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 3, 10, 14, and 17 are rejected under 35 U.S.C. 103(a) as being unpatentable over by Arslan et al. (US. Patent No. 6,263,307) in view of Im et al. (US. Patent No. 5,805,696).

8. Arslan et al. discloses all the limitations of claim 3, but fails to specifically disclose a coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient. However, Im et al. teaches a coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient (figures 2 or 3). The advantage using the method taught by Im et al. is to recursively calculate and update the filter coefficient to appropriately reduce the adaptive noise.

Since Arslan et al. and Im et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by incorporating an adaptive filter coefficient calculation unit as taught by Im et al. in order to calculate and update the filter coefficients to appropriately reduce the adaptive noise that corrupts the input signal.

9. Arslan et al. discloses all the limitations of claim 10, but fails to specifically disclose a coefficient calculation unit executes an adaptive control to a signal having an

input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient. However, Im et al. teaches a coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient (figures 2 or 3). The advantage using the method taught by Im et al. is to recursively calculate and update the filter coefficient to appropriately reduce the adaptive noise.

Since Arslan et al. and Im et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by incorporating an adaptive filter coefficient calculation unit as taught by Im et al. in order to calculate and update the filter coefficients to appropriately reduce the adaptive noise that corrupts the input signal.

10. Arslan et al. discloses all the limitations of claim 14, but fails to specifically disclose a coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient. However, Im et al. teaches a coefficient calculation unit executes an adaptive control to a signal having an input voice signal and a simulated voice signal added, and obtains a tap coefficient to thereby calculate the filter coefficient (figures 2 or 3). The advantage using the method taught by Im et al. is

to recursively calculate and update the filter coefficient to appropriately reduce the adaptive noise.

Since Arslan et al. and Im et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by incorporating an adaptive filter coefficient calculation unit as taught by Im et al. in order to calculate and update the filter coefficients to appropriately reduce the adaptive noise that corrupts the input signal.

11. Referring to claim 17, Arslan et al. discloses a method of extracting voice features, wherein the noise reduction system coefficient is attained by applying a fast Fourier transform to the filter coefficient to thereby calculate the power spectrum vector (902 of figures 9a or 9b). Arslan et al. fails to specifically disclose an adder for adding a surrounding voice signal received by the microphone and a specific simulated voice signal and executing an adaptive control to the added signal to thereby calculate a filter coefficient.

However, Im et al. teaches an adder for adding a surrounding voice signal received by the microphone and a specific simulated voice signal (30 of figure 3) and executing an adaptive control to the added signal to thereby calculate a filter coefficient (CONTROL of figure 3). The advantage of using the method taught by Im et al. is to recursively calculate and update the filter coefficient to appropriately reduce the adaptive noise.

Since Arslan et al. and Im et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by incorporating an adder for adding the two signal together and executing adaptive control to calculate the filter coefficient as taught by Im et al. in order to calculate and update the filter coefficients to appropriately reduce the adaptive noise that corrupts the input signal.

Claims 5-7 are rejected under 35 U.S.C. 103(a) as being unpatentable over Arslan et al (US. Patent No. 6,263,307) in view of LaRue (US. Patent No. 5,274,560).

12. Referring to claim 5, Arslan et al. discloses all the limitations of claim 5, but fails to specifically disclose a voice feature extraction device, wherein the voice feature extraction device is applied to a voice recognition device of a vehicle navigation system. However, LaRue teaches a voice feature extraction device, wherein the voice feature extraction device is applied to a voice recognition device of a vehicle navigation system (figure 4, subsystem 4 extracts speech features). The advantage of using the method taught by LaRue is to allow drivers communicate verbally with the voice recognition navigation system without taking off an eye from the traffic.

Since Arslan et al. and LaRue are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by applying the voice feature extraction to a voice recognition device of a vehicle navigation system as taught by

LaRue in order to analyze the input speech command to appropriately retrieve direction and verbally guide the driver so that the driver can pay attention to the traffic.

13. Referring to claim 6, Arslan et al. discloses all the limitations of claim 6, but fails to specifically disclose a voice feature extraction device, wherein the voice feature extraction device is applied to a speaker recognition device. However, LaRue teaches a voice feature extraction device, wherein the voice feature extraction device is applied to a speaker recognition device (col. 6, ln. 32-44).

Since Arslan et al. and LaRue are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by applying the voice feature extraction to the speaker recognition device as taught by LaRue in order to train the speaker recognition device with different spoken accents to further enhance the recognition capabilities of the system and make it more reliable.

14. Referring to claim 7, Arslan et al. discloses all the limitations of claim 7, but fails to specifically disclose a voice feature extraction device a voice feature extraction device, wherein the voice feature extraction device is applied to a loudness compensation system. However, LaRue teaches a voice feature extraction device, wherein the voice feature extraction device is applied to a loudness compensation system (col. 6, ln. 21-26, a system that plays music is considered a loudness

compensation system). The advantage of using the method taught by LaRue is to allow music listeners to control speakers' volume by verbal communications.

Since Arslan et al. and LaRue are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by applying the voice feature extraction to the speaker recognition device as taught by LaRue in order to train the speaker recognition device with different spoken accents to further enhance the recognition capabilities of the system and make it more reliable.

Claims 4, 11, 15, and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over by Arslan et al. (US. Patent No. 6,263,307) in view of Im et al. (US. Patent No. 5,805,696) and further in view of Haykin et al. (US. Patent No. 5,027,123).

15. Referring to claim 4, the combination of Arslan et al. and Im et al. discloses all the limitations of claim 4, but fails to specifically disclose a voice feature extraction device, wherein a specific gain adjustment is executed on the simulated voice signal. However, Haykin et al. teaches voice feature extraction device, wherein a specific gain adjustment is executed on the simulated voice signal (33 of figure 4). The advantage of using the method taught by Haykin et al. is to recursively calculate and update the filter coefficient at various levels of the simulated voice signal measure the system's reliabilities.

Since the modified Arslan et al. and Haykin et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Arslan et al. by incorporating an adaptive filtering coefficient calculation unit as taught by Haykin et al. in order to calculate and update the filter coefficients to appropriately reduce adaptive noise that corrupts the input signal.

16. Referring to claim 11, the combination of Arslan et al. and Im et al. discloses all the limitations of claim 11, but fails to specifically disclose a voice feature extraction device, wherein a specific gain adjustment is executed on the simulated voice signal. However, Haykin et al. teaches voice feature extraction device, wherein a specific gain adjustment is executed on the simulated voice signal (33 of figure 4). The advantage of using the method taught by Haykin et al. is to recursively calculate and update the filter coefficient at various levels of the simulated voice signal measure the system's reliabilities.

Since the modified Arslan et al. and Haykin et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Arslan et al. by incorporating an adaptive filtering coefficient calculation unit as taught by Haykin et al. in order to calculate and update the filter coefficients to appropriately reduce the adaptive noise that corrupts the input signal.

17. Referring to claim 15, the combination of Arslan et al. and Im et al. discloses all the limitations of claim 15, but fails to specifically disclose a voice feature extraction device, wherein a specific gain adjustment is executed on the simulated voice signal. However, Haykin et al. teaches voice feature extraction device, wherein a specific gain adjustment is executed on the simulated voice signal (33 of figure 4). The advantage of using the method taught by Haykin et al. is to recursively calculate and update the filter coefficient at various levels of the simulated voice signal measure the system's reliabilities.

Since the modified Arslan et al. and Haykin et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Arslan et al. by incorporating an adaptive filtering coefficient calculation unit as taught by Haykin et al. in order to calculate and update the filter coefficients to appropriately reduce the adaptive noise that corrupts the input signal.

18. Referring to claim 18, Arslan et al. discloses a voice feature extraction device comprising:

a microphone that receives a surrounding voice signal (figures 1a or 1b);

an FFT operation unit that executes a fast Fourier transform to a filter coefficient obtained by the adaptive control of the adaptive filter (figure 9b);

a power calculation unit that calculates a power spectrum vector from a signal calculated by the FFT operation unit (904 figure 9b); and

a noise reduction system having the power spectrum vector calculated by the power calculation unit set as a noise reduction coefficient (figure 9b).

Arslan et al. fails to specifically disclose a simulated voice signal generation unit that generates a specific simulated voice signal; a gain adjustment unit that adjusts a gain of the simulated voice signal; an adder that adds the voice signal received by the microphone and the gain-adjusted simulated voice signal; a delay processing unit that delays the gain-adjusted simulated voice signal by a specific time; and an adaptive filter that executes an adaptive control on the basis of the signal added by the adder and the simulated voice signal delayed by the delay processing unit.

However, Im et al. teaches a simulated voice signal generation unit that generates a specific simulated voice signal (34 of figure 3); an adder that adds the voice signal received by the microphone and the gain-adjusted simulated voice signal (30 of figure 3); and an adaptive filter that executes an adaptive control on the basis of the signal added by the adder and the simulated voice signal delayed by the delay processing unit (20 of figure 3). The overall advantage of using the operation taught by Im et al. mentioned above is to recursively calculate and update the filter coefficient so that adaptive noise can be removed appropriately.

Since Arslan et al. and Im et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Arslan et al. by incorporating a simulated voice signal unit, and adder unit, and an adaptive filter as taught by Im et al. in order to

calculate and update the filter coefficients to appropriately reduce the adaptive noise corrupts the input signal.

The combination of Arslan et al. and Im et al. still fails to disclose a gain adjustment unit that adjusts a gain of the simulated voice signal and a delayed processing unit that delays the gain-adjusted simulated voice signal by a specific time. However, Haykin et al. teaches a gain adjustment unit that adjusts a gain of the simulated voice signal (33 of figure 4) and a delayed processing unit that delays the gain-adjusted simulated voice signal by a specific time (elements 27, 31, 33, 35, and 37 of figure 4 together can form a delay unit). The overall advantage of using the method taught by Haykin et al. mentioned above is to recursively calculate and update the filter coefficient at various levels of the simulated voice signal to measure the system's reliabilities.

Since the modified Arslan et al. and Haykin et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Arslan et al. by incorporating a gain adjustment unit and a delayed processing unit as taught by Haykin et al. in order to control the simulated voice signal to achieve a precise calculation of filter coefficients.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Huyen Vo whose email address is vo.huyen@uspto.gov. The examiner can normally be reached on M-F, 9-5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Doris To can be reached on 703-305-4827. The fax phone number for the organization where this application or proceeding is assigned is (703) 872-9306.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is 703-306-0377.

Examiner: Huyen X. Vo

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October 23, 2003

11/14/03

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